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**SUBSTITUTE SPECIFICATION**

## SPECIFICATION

## TITLE

## "METHOD AND DEVICE FOR ADAPTIVELY MODIFYING THE CHARACTERISTICS OF ONE-DIMENSIONAL SIGNALS"

## BACKGROUND OF THE INVENTION

### Field of the Invention

The present invention is directed to a signal processing method as well as to an apparatus for the realization of this method.

### Description of the Prior Art

In technology, information-carrying time curves of individual physical quantities (for example, acoustic pressure fluctuations given an acoustic signal) must often not only be amplified but also modified in terms of their properties. Such curves are referred to as one-dimensional signals. In general, the desired modification of the properties of such signals is referred to herein as feature modification. Due to the practical advantages connected therewith, this feature modification currently usually ensues with electronic means. This requires the signal to be modified in terms of its properties to represent an electrical voltage fluctuation. It is converted into such a voltage fluctuation with a suitable transducer (for example, by a microphone given an acoustic signal). After the feature modification, the signal must often be converted back into the original physical form of representation, a suitable transducer being required for this purpose (for example, a speaker given an acoustic signal). Digital methods are often employed for the feature modification because these exhibit a number of well-known advantages.

Often, a feature modification that is well-known and that is referred to as frequency-selective filtering is applied. Certain parts of the signal spectrum are thereby emphasized, others attenuated. Another known group of feature modifications is referred to as optimum filtering or noise-reducing filtering. These involve recognizing parts of the signal that derive from noise sources or other unwanted signal paths (for example, acoustic feedback paths) and separating them from the signal such that the desired payload signal is recovered optimally unfalsified. Another known group of feature modifications is referred to as compression. The signal amplitudes are attenuated to a greater or lesser extent dependent on their momentary intensity. Examples are the transmission of a high-dynamics signal via a radio channel with limited modulation amplitude or the adaptation of a voice signal to the reduced range of dynamics of impaired hearing with a suitable hearing aid device such as disclosed, for example, by United States Patent No. 3,894,195 (K.D.Kryter). Another form of feature modification is the temporary boosting of specific signal parts known as pre-emphasis, with the goal of making the signal more resistant to the influence of noise incursions in a transmission channel. Finally, the boosting of specific parts of a signal, with the goal of producing the sensation of an especially pleasant sound for the hearer, is also to be viewed as feature modification in the above sense.

When the ambient conditions change, the desired processing result can often be achieved or maintained in an optimum way only when the described feature modifications modify their characteristic in time-varying

fashion, i.e. adapt to the changing ambient conditions. Numerous strategies for the adaptation of the described feature modifications are known. These are not the subject matter of this invention but such changing conditions have an unavoidable influence on the quality of the processing.

The adaptive "Overlap-Add" method recited by J. B. Allen and L. Rabiner in "A unified approach to short-time Fourier analysis and synthesis", which appear in Proc. of the IEEE, Vol. 65, 1977, pages 1558-1564, has proven itself especially for the implementation of the described feature modifications. The calculating outlay of the method is still low even when extremely complex feature modifications are required. Moreover, the adaptation (not described in detail here) to changing ambient conditions is supported by the "Overlap-Add" method because it already contains the calculation of short-time estimated values of the signal spectrum that exhibit high statistical dependability. In the "Overlap-Add" method, the input signal digitized in the analog-to-digital converter is continuously subdivided into blocks with the same number M of samples that overlap one another. Each block is multiplied by a suitable window in order to maximize the estimation precision of the subsequent transformation. An estimated value for the spectrum of the segment is calculated from each block with a fast Fourier transform (FFT), whereby the transformation length N must be greater than the block length M, as shall be substantiated below. The feature modification ensues by multiplying the n spectral values of each data block with suitably selected weighting factors. The back-transformation (inverse

fast Fourier transform, IFFT) supplies a block of the modified output signal. After the superimposition of successive blocks, now a signal that is again a continuous output signal can be further-employed (thus, for example, also converted back into the original physical form of representation).

What has proven disadvantageous, however, is that the output signal is very often affected with errors that arise in the IFFT of the weighted spectra. It is known that the IFFT of N values of a spectrum supplies the basic period, likewise having the length N, of a periodic time sequence (see, for example, A. V. Oppenheim, R. W. Schafer, Zeitdiskrete Signalverarbeitung, Munich: Oldenbourg, 1995). When the spectrum was generated from a time signal by an FFT of the same length N, then the result of the IFFT corresponds exactly to the input signal. When, in contrast, the spectrum was multiplicatively weighted, as described above, then this corresponds to a convolution with a filter pulse response (having a generally unknown length L) in the time domain. The result of such an operation, as is known, has a length that nearly corresponds to the sum of the input signal block length m and the length L of the pulse response. The basic period with the length N used after the IFFT for the reconstruction of the output signal, however, is then an excerpt from an additive superimposition of infinitely many repetitions of the excessively long convolution result respectively shifted by N samples (referred to as circular convolution), as likewise explained in the aforementioned book by Oppenheim and Schafer. These shifted superimposed signal values are audible as errors in the output signal

(what is referred to as time domain aliasing). In order to at least diminish these errors, it has been proposed to select the length of FFT and IFFT significantly greater than the input signal block length M in order to obtain optimally few superimpositions within the basic period of the length N. As a result, the calculating outlay is drastically increased without a dependable error limitation being possible. The application of a further window on the output data blocks supplied by the IFFT, which has already been attempted, also allows no dependable error limitation but, on the other hand, leads to further signal falsifications.

The errors can be dependably avoided in that the length L of the filter pulse response, which corresponds to the weighting factors, is suitably limited. Since, however, these weighting factors are only supplied and continuously adapted in the frequency domain in a great number of adaptive feature modification methods, a filter design method would have to be implemented after every change, for example according to the strategy of frequency sampling (see, for example, the aforementioned book by Oppenheim and Schafer). This requires a great deal of calculating time, would thus interrupt the continuous processing and would make the real-time application of most of the aforementioned feature modification methods impossible.

## SUMMARY OF THE INVENTION

An object of the present invention is to provide a method and a device for adaptive feature modification of one-dimensional signals which avoids the aforementioned problems associated with known methods and devices.

This object is achieved in accordance with the principles of the present invention in a method and device wherein an incoming analog one-dimensional signal is converted into a digital signal using the adaptive overlap-add algorithm, wherein the feature modification takes place after a discrete spectral transformation by multiplication in the frequency domain with a frequency response function, and wherein the output signal is subsequently generated by a corresponding inverse discrete spectral transformation as well as by overlapping and shifted addition of a number of signal segments which are produced by the inverse spectral transformation, and wherein, before the multiplication in the frequency domain, the frequency response function is convolved (convoluted) with a selected, discrete window function that has a significantly shorter length than the frequency response function.

The introduction of the supplemental algorithm into the adaptive overlap-add method allows the time domain aliasing errors to be dependably kept below a selectable limit value. The calculating outlay for this supplemental algorithm is slight; in most instances, it only amounts to a small fraction of the outlay that is necessary anyway for the overlap-add method. It has also proven advantageous that the calculating outlay of the inventive

supplemental algorithm decreases when greater errors are allowed. As a result, an optimum compromise between calculating outlay and quality of the output signal can be found for every application. The concept and the application of the inventive error limitation method are explained by way of example below on the basis of the Figures without other embodiments that are obvious to persons knowledgeable in the technology being thereby precluded or limited.

## DESCRIPTION OF THE DRAWINGS

Figure 1 is a block circuit diagram of a known device for feature modification of one-dimensional signal, operating according to a known method for feature modification.

Figure 2 shows an exemplary strategy for the reconstruction of the output signal by overlapping addition of signal segments.

Figure 3 is a block circuit diagram of a device for feature modification of one-dimensional signal constructed and operating in accordance with the principles of the present invention.

Figure 4 illustrates the operation of the shift registers in the inventive device, with exemplary frequency response and window sequences being shown therein.

Figure 5 illustrates the result of the back-transformation of the window function shown in Figure 4 into the time domain, in accordance with the principles of the present invention.

## DESCRIPTION OF THE PREFERRED EMBODIMENTS

Figure 1 shows the block circuit diagram of a known apparatus for feature modification. It is thereby assumed that the input signal x(t), which is brought to the analog-to-digital converter 20 via the line 10, represents a continuous electrical voltage curve over the time t. In certain applications, this signal was generated in a known way from the curve of another physical quantity (for example, acoustic pressure) using a suitable transducer (for example, microphone. The block 20 (analog-to-digital converter) also performs the sample-and-hold function when such a function is required for the conversion method. It is assumed that the values are quantized with adequate precision, i.e. an adequate number of bits per value x(n). Via the line 30, thus, a time sequence x(n) of digitalized sampled of the input signal is brought to the shift register 40 and to the processing unit 50, which realizes the adaptation strategy. The index n represents the count index of the successive sampling intervals. Optionally, the transmission can ensue parallel or serially; this is of no consequence for the processing method to be explained here. M samples x(n) are stored in the shift register 40. When a new value is read in, the oldest stored value is shifted out and is lost. A processing cycle begins when K new values are read in. M must be a whole multiples of K; M=K is possible but not expedient. M samples are taken from the shift register 40 and supplied to a multiplier arrangement 70 via the line 60. The line 60 is shown as a ribbon of M parallel, discrete lines in order to symbolize that a block of M samples is processed in respectively one

processing step. In the multiplier arrangement 70, each input value is multiplied by respectively one value of a window function w(n) that is kept ready in a memory 80. The block 70 can contain M multipliers that operate in parallel. However, one multiplier can suffice when it is fast enough to handle M multiplications in time-division technique within k sampling intervals. The window function w(n) must be selected such that it supplies a constant sequence given shifting by multiples of K and summation, as explained in greater detail in the aforementioned publication of Allen and Rabiner.

The M window-weighted input data are supplemented by N-M zero values, this being symbolized by the arrangement 90. The data block of N values that has arisen in this way is supplied to a discrete spectral transformation 100. Allen and Rabiner proposed the discrete Fourier transformation in the calculation-efficient embodiment of fast Fourier transform (FFT) for this. The spectral transformation supplies a spectrum X(v) of N discrete, often complex values that are supplied to a multiplier arrangement 120 via the line 110. The feature modification ensues in the multiplier arrangement 120 by every value X(v) being multiplied by a value of the frequency response function H(v) stored in the memory 130. A discrete output spectrum

$$Y(v) = X(v) \cdot H(v) \qquad \text{for } v = 0,1,2,\ldots,N-1 \qquad (1)$$

arises.

From time to time, the adaptation strategy (symbolized by the block 50) modifies individual or all values of H(n) in the memory 130 via the line 140 dependent on variations in the input signal x(n). This is not described in greater detail here since the adaptation strategy is not the subject matter of the present invention.

The output spectrum Y(n) is supplied via a line 150 to an inverse spectral transformation 160 that represents the inversion of the transformation employed in the block 100, so that N digital samples $y_i(n)$ of a time domain signal are calculated. The index i is the running index of the processing steps. As described above, only K new samples but M-K older samples, which were already employed once or even repeatedly in earlier processing steps, are supplied to the processing from the shift register in every processing step. The processing thus ensues in overlapping data sections. Accordingly, the final result function y(n) arises at the output by time-shifted summation of a number of signal segments $y_i(n)$. This is accomplished by the adder arrangement 170, the two shift registers 180 and 190 as well as the switchover unit 200. The adder arrangement 170 implements N-k additions in every processing step. It is irrelevant whether N-K parallel adders are realized or whether one fast adder handles the additions in a time-division multiplex method. The results of the additions and the K newest result values that have not yet been subjected to the addition are entered in parallel into the shift register 180 and are then serially shifted out toward the top. The switchover unit 200 is initially placed toward

the right, so that the K oldest values stored in the shift register are supplied to the digital-to-analog converter 210 and thus form a part of the output sequence y(n). The switchover unit 200 is then switched to the left and the remaining N-K values are transferred into the shift register 190, so that these are available for further additions.

Figure 2 shows the time-shifted and overlapping summation of the signal segments $y_i(n)$, i=0,1,2,..., to form the output sequence y(n0 for the exemplary values N=256 and K=64.

The digital-to-analog converter 210 generates a continuous electrical output signal y(t) from y(n). This can be supplied for further-processing via the line 220, for example to a conversion into another physical form of representation.

The processing shown in Figure 1, however, ensues free of time domain aliasing errors only when a pulse response h(n) in the time domain, that comprises no more than L values differing from zero belongs to the frequency response function H(v) present in the memory 130, L satisfying the condition

$$L \leq N-M+1 \qquad (2).$$

If the adaptation strategy 50 continuously offers a new frequency response function H(v), the check of the condition (2) is so involved that real-time processing is generally no longer possible, as was already pointed out above.

Figure 3 shows the block circuit diagram of an apparatus for feature modification in accordance with the invention. Figure 3 differs from Figure 1 in that the frequency response function H(v), after being defined by the adaptation strategy 50, is first read in parallel into a shift register 131 having the length N via the line 140. When shifted, the output values of the shift register 131 are written back into the input cell of the same shift register via the return line 132 and are also supplied to a multiplier 133. Another shift register 134 contains a suitable window function G(v) having what is generally a short overall length J. When the values G(v) are shifted upward out of the shift register 134, then these are written back into the input cell of the same shift register 134 via the return line 135. The multiplier 133 multiplies the values respectively pending at outputs of the shift registers 131 and 134 and supplies the product to an adder 136. The addition 136 adds the value pending at the output of the memory 137 thereto and overwrites the previous value in the memory 137 with the sum. The adder 136 and the memory 137 thus form an accumulator. When a new accumulation is to be begun, the memory 137 must be erased, which is symbolized by the line 138. It is known from the literature, for example the book "Digitale Signalverarbeitung, Volume 1, by H.W.Schüßler, Springer-Verlag, Berlin (4$^{th}$ Edition, 1994), that the arrangement of the shift registers 131 and 134, the multiplier 133, the adder 136 and the memory 137 forms a non-recursive digital filter. This implements the circular convolution operation insofar as the discrete function contained in the shift register 134 was written in the

opposite direction. This latter is not required here because convolution is always carried out with only a window function that represents a symmetrical sequence. Since the length J of the window function G(v) is generally significantly shorter here than the length of the frequency response H(v), the following executive sequence of the convolution derives: The window function G(v) is stored in the shift register 134 shifted by B values of the frequency index v, whereby

$$B = \begin{cases} \frac{J}{2} \text{ given even } J \\ \frac{J-1}{2} \text{ given odd } J \end{cases} \tag{3}$$

applies. Figure 4 shows this as an example for J=9. One can see that the value G(-B), G(-4) here, resides in the output cell of the shift register 134 before the beginning of the convolution operation. After the parallel read-in of a new frequency response H(v) into the shift register 131 via the line 140, this is first to be likewise shifted by B values of the index v, whereby B is established by Equation (3). Figure 4 also shows this as an example for an unrealistically small but surveyable value N=16, whereby real values of H(v) were assumed for simplification. As the Fourier transform of a time-discrete pulse response, H(v) exhibits periodicity with regard to v with the period N; the values of v residing next to one another in Figure 4 and separated by a slash are therefore both equally applicable. The shift of H(v) shown in Figure

4 occurs when the shift register 131 initially executes N-B shift steps (given a stationary shift register 134 and the multiplier 133 deactivated). The memory 137 is then erased. The values H(-B) and G(-B) now pending at the shift register outputs are multiplied by the multiplier 133, the adder 136 adds zero, so that the product proceeds into the memory 137 unmodified. Both shift registers 131 and 134 are now shifted once. The multiplier 133 forms the product H(-B+1) · G(-B+1). This is added in the adder 136 to the stored product H(-B) · G(-B+1) and the sum is stored in the memory 137. One continues in this way until J partial products H(v) · G(v) are added and the overall sum is stored in the memory 137. The first value of the frequency response $\tilde{H}(0)$ modified by the convolution now pends at the output of the memory 137. This value is written in the shift register 130 by means of a one-time shift of the memory 130, fashioned as a shift register here. The multiplier 133 and the shift register 134 are subsequently stopped, whereas the shift register 131 executes another N-J+1 shift steps. Since the beginning of the convolution operation, the latter register has thus implemented N+1 shift steps, as a result of which a shifting of H(v) that is smaller by a value than shown in Figure 4 occurs, i.e. H(-B+1), H(-3) here, now resides in the output cell of the shift register 131. When the memory 137 is subsequently erased and the product H(-B+1) · G(-B) is subsequently written into the memory 137, then this is the first step for calculating the second modified frequency response value $\tilde{H}(1)$. When the calculation of this value has been completed, this value is also shifted from the memory

137 into the shift register 130. This is continued until N modified frequency response values H(v) ultimately reside in the shift register 130. From there, these can be repeatedly transmitted in parallel into the multiplier arrangement 120 in order to effect the feature modification in the input data blocks in the way that was already described above. Only when the adaptation strategy 50 determines that a new frequency response function H(v) must be applied is the inventive windowing re-activated.

The exemplary device for dynamics compression of voice signals realized according to Figure 3 can be realized small, lightweight and, therefore, easily worn as an experimental hearing aid when the parameters are specified as follows: input signal segment M=180; plurality of new values in each input signal segment: K=90; fashioning the transformation as FFT with the length N=256; length of the window G(v): J=9. In the realization with the assistance of assembler code of a modern, fast signal processor, for example the DSP56L002 of Motorola Semiconductor Ltd., the calculating outlay of the inventive device (according to Figure 3) proved to be only slightly increased, namely by approximately 12%, compared to a device manufactured according to the conventional Prior Art and corresponding to Figure 1.

It can be concluded from theoretical treatises, for example in "the digital prolate spheroidal window" by T. Verma, S. Bilbao and T. H. Y. Meng, which appeared in the Proceedings of the International Conference on Acoustics, Speech and Signal Processing (ICASSP) 1996, presented by the

IEEE in Atlanta, USA, pages 1351-1354, that the least remaining time domain aliasing errors were achieved with the prolate spheroidal window function. Figure 4 shows such a window function G(v) with the length J=9 in the shift register symbol 134. Figure 5 shows the appertaining time domain window function g(n) as obtained after an inverse FFT having the length 256. According to the convolution theorem (explained in the cited book by Oppenheim and Schafer), the convolution of the frequency response H(v) with the window function G(v) in the time domain corresponds to the multiplicative window weighting of h(n), the pulse response belonging to H(v), by g(n). Figure 5 shows that the pulse response cannot be made exactly time-limited by a window function selected according to patent claim 2. However, the values of the pulse response that do not lie in the region of the principal lobe of the window g(n) according to Figure 5 are multiplied by very small factors and thus largely neutralized. It therefore turns out that the time domain aliasing errors were already capable of being reduced by the very short window function shown in Figure 4 to below 1% of those errors caused by a device corresponding to the conventional Prior Art. Technically experienced persons know from the literature and from experience that an enlargement of the length J of the prolate spheroidal window further reduces the side lobes of the time domain window g(n). The time domain aliasing errors are thereby also reduced further. Together with J, however, the calculating outlay is also increased, as the above description of the convolution operation shows. A beneficial compromise between the size of

the remaining processing errors and the calculating outlay can thus be easily found.

The spatial transformation and the inverse spectral transformation can be a discrete Haar transformation and an inverse Haar transformation, or a discrete Walsh. Hadamard transformation and an inverse Walsh-Hadamard transformation, or a discrete Hartley transformation and an inverse Hartley transformation, or a discrete cosine transformation and an inverse cosine transformation.

Although modifications and changes may be suggested by those skilled in the art, it is the intention of the inventors to embody within the patent warranted hereon all changes and modifications as reasonably and properly come within the scope of their contribution to the art.